Abstract

A digital audio signal to be replayed is processed in a waveform thereof. A frequency bandwidth of the audio signal is expanded through conversion of a sampling frequency, and then the audio signal is low-pass-filtered with a low-pass cut-off frequency corresponding to the converted sampling frequency. An interval of time between two waveform peaks of the audio signal is detected, and then difference data between current data of the audio signal and past data thereof is calculated. The difference data are subject to weighting depending on the interval, and then output data are produced based on both the low-pass-filtered audio signal and the weighted difference data. This processing, which can be realized by activation of software, improves audio quality when compressed audio data is replayed.